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EXAMINER	
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)
	10/803,420	SINGHAL ET AL.
	Examiner	Art Unit
	MICHAEL C. COLUCCI	2626

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 27 March 2009.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-18 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-18 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

1. Applicant's arguments filed 03/27/2009 have been fully considered but they are not persuasive.

Argument (page 9 paragraph 2):

- “Chen clearly fails to teach "applying a window function to the remaining frames," as set forth in Applicant's independent claims 1 and 6, and "the at least one controller configured to apply a window function to the remaining frames," as set forth in Applicant's independent claim 11. Thus, the combination of Oh and Chen fail to disclose "apply[ing] a window function to the remaining frames," as set forth in Applicant's independent claims 1, 6 and 11”

Response to argument:

Oh in view of Chen explicitly teach skipping frames based on a playback speed, wherein Oh particularly teaches well uses of playback speed and the skipping of frames (Fig. 1). In light of the specification (present invention spec. [0008]), Examiner maintains the use of Oh, wherein Oh describes the use of a window function applied to as speech signal. However, Examiner has incorporated Chen to explicitly describe the intended and yet well known use of a window function on an audio signal in particular. Chen describes that which is well known in the art. Consider that the inherency of a window function is directed to preserving signal data within a window/interval, wherein any data outside the interval is

zeroed (or muted for an audio signal, and thus skipped). The concept of Oh is realized through the teaching of Chen, wherein by applying a window function to frames, the zeroing or muting frames is present. Also, consider that a window function itself can be applied to a speech signal with given parameters that allow for the elimination of data outside a windowed frame (i.e. other frames NOT in the window). Chen thus teaches attenuation of the signal outside the windowed area (Chen Col. 9 lines 9-38). Further, consider the purpose of a window function in a speech signal in the instance where increasing the playback speed alone may have residual undesirable effects. Thus the use of a window function as taught by Chen, would alleviate any residual effects or noise present after skipping frames. This is also consistent with the present invention, wherein both Chen and the present invention teach the concept of overlapping as well as "smoothing" a signal out (present invention [0028]). Chen in explicitly teaches the elimination of errors (residual effects or noise) by the well known use of a window function to completely stop any surrounding errors. Chen also differentiates between the use of partial and full attenuation through window functions (Chen Col. 10 lines 1-26). Thus it would have been obvious to combine the well known teachings of Chen's window function to the playback speed variation of Oh to allow for the guaranteed elimination of any residual signal even when frames are skipped.

Claim Rejections - 35 USC § 103

2. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

3. Claims 1, 4-6, 9-11, and 14-18 are rejected under 35 U.S.C. 103(a) as being unpatentable over Oh et al. US 5781696 (hereinafter Oh) in view of Chen et al. US 6915263 B1 (hereinafter Chen).

Re claims 1, 6, and 11, Oh teaches a method for speeding up an encoded original audio signal, said original audio signal having an original frequency and original playback speed, said method comprising:

retrieving frames of the original audio signal (Fig. 5);

wherein said desired playback speed is greater than the original playback speed (col. 5 lines 60-65);

applying a window function (col. 5 line 65 – col. 6 line 2) to the remaining frames

converting the signal with the windowed frames from digital to analog format;

using the original frequency to playback the analog format signal (col. 6 lines 38-46)

skipping frames at a rate according to a desired playback speed (Col. 1 lines 33-45, every other frame at a higher play back speed);

However, Oh fails to teach receiving the encoded original audio signal;

applying a window function (col. 5 line 65 – col. 6 line 2) to the remaining frames

Chen teaches error reduction of encoded frames, wherein Chen teaches error entries of error array 370 can be computed and stored by the parser process 270 of the decoder 200. There are several ways in which the AC3 data can indicate that errors are contained within a frame of encoded data. In one method, the decoder 200 can be informed of the error frame by the transport system which delivers the data. The data integrity can also be checked using the embedded CRC 220 fields for each encoded frame. Methods for using the CRC fields of an encoded frame for error detection are well known. Also, well known consistency checks on the received bitstream 134 can also be used to indicate that errors are present in a particular encoded frame. It is appreciated that at step 305 of FIG. 4, any of a number of well known processes can be used for generating the error array 370 of FIG. 5A based on the input bitstream 134. In the example of FIG. 5A, the next audio encoded frame that is being processed at step 305 is frame 48. (Chen Col. 7 lines 37-55).

Further, Chen teaches well known techniques in playback processing of skipping a current frame and the output being muted (whether or not the current frame contains an error therein), otherwise, the current frame is normally decoded and played. In this way, the number of transition times from normal play to mute and from mute to normal play (unmute) is reduced. In effect, the muting strategy is extended across several non-error frames depending on the accumulated error rate so that short mutings are merged into a long muting. When the error rate is high, process 280 acts to merge together

adjacent error frames (mute merging) by increasing the error recovery delay period. The amount of mute merging is adaptive and is based on the error rate. At step 345, a number of different muting operations can be performed to mute the current frame. In the preferred embodiment, a smooth muting with zeros can be applied to decline the audio signal at a given rate according to a window function and in an alternate embodiment, a frame repeat can be performed. FIG. 6 illustrates smooth muting with zeros to reduce the "pop" sounds associated with muting. In this embodiment, an attenuation or "window" function 420 is applied to the decoded audio frame represented as signal 410 to decline its amplitude. Windowing starts at the zero-cross point. The attenuation function represents the amount of the original signal 410 allowed to exist at any given time and the remainder of the audio signal is padded (e.g., replaced) with zeros to provide a mute. Smoothing functions and muting using window functions are well known (Col. 9 lines 9-38).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Oh to incorporate receiving the encoded original audio signal and applying a window function as taught by Chen to allow for the smoothing of a signal after certain frames were removed/muted, wherein a windowing function is applied to frames when skipping or muting frames if an error occurs prior to processing (Col. 9 lines 9-38), and to allow for the guaranteed elimination of any residual signal even when frames are skipped (Chen Col. 10 lines 1-26).

Re claims 4, 9, and 14, Oh teaches the method according to claim 1 wherein the desired playback speed is a predefined default value (col. 6 lines 34-38).

Re claims 5, 10, and 15, Oh teaches the method according to claim 1 wherein the desired playback speed is a programmable value (col. 6 lines 34-38).

Re claims 16-18, Oh fails to teach the method of claim 1, wherein skipping frames at a rate according to a desired playback speed further comprises skipping frames at a rate according to a desired playback speed, wherein the frames correspond to time intervals (Col. 1 lines 33-45, every other frame at a higher play back speed);.

Chen teaches error reduction of encoded frames, wherein Chen teaches error entries of error array 370 can be computed and stored by the parser process 270 of the decoder 200. There are several ways in which the AC3 data can indicate that errors are contained within a frame of encoded data. In one method, the decoder 200 can be informed of the error frame by the transport system which delivers the data. The data integrity can also be checked using the embedded CRC 220 fields for each encoded frame. Methods for using the CRC fields of an encoded frame for error detection are well known. Also, well known consistency checks on the received bitstream 134 can also be used to indicate that errors are present in a particular encoded frame. It is appreciated that at step 305 of FIG. 4, any of a number of well known processes can be used for generating the error array 370 of FIG. 5A based on the input bitstream 134. In

the example of FIG. 5A, the next audio encoded frame that is being processed at step 305 is frame 48. (Chen Col. 7 lines 37-55).

Further, Chen teaches well known techniques in playback processing of skipping a current frame and the output being muted (whether or not the current frame contains an error therein), otherwise, the current frame is normally decoded and played. In this way, the number of transition times from normal play to mute and from mute to normal play (unmute) is reduced. In effect, the muting strategy is extended across several non-error frames depending on the accumulated error rate so that short mutings are merged into a long muting. When the error rate is high, process 280 acts to merge together adjacent error frames (mute merging) by increasing the error recovery delay period. The amount of mute merging is adaptive and is based on the error rate. At step 345, a number of different muting operations can be performed to mute the current frame. In the preferred embodiment, a smooth muting with zeros can be applied to decline the audio signal at a given rate according to a window function and in an alternate embodiment, a frame repeat can be performed. FIG. 6 illustrates smooth muting with zeros to reduce the "pop" sounds associated with muting. In this embodiment, an attenuation or "window" function 420 is applied to the decoded audio frame represented as signal 410 to decline its amplitude. Windowing starts at the zero-cross point. The attenuation function represents the amount of the original signal 410 allowed to exist at any given time and the remainder of the audio signal is padded (e.g., replaced) with zeros to provide a mute. Smoothing functions and muting using window functions are well known (Col. 9 lines 9-38).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Oh to incorporate skipping frames at a rate according to a desired playback speed further comprises skipping frames at a rate according to a desired playback speed, wherein the frames correspond to time intervals as taught by Chen to allow for the smoothing of a signal after certain frames were removed/muted, wherein a windowing function is applied to frames when skipping or muting frames if an error occurs prior to processing (Col. 9 lines 9-38).

4. Claims 2, 3, 7, 8, 12, and 13 are rejected under 35 U.S.C. 103(a) as being unpatentable over Oh et al. US 5781696 (hereinafter Oh) in view of Chen et al. US 6915263 B1 (hereinafter Chen) and further in view of Kizuki et al. US 5684829 A (hereinafter Kizuki).

Re claims 2, 7, and 12, Oh in view of Chen fails to teach the method according to claim 1 wherein the encoded original audio signal is encoded in the frequency domain using one of a plurality of encoding schemes, the method further comprising frequency-domain decoding of the encoded original audio signal.

Kizuki teaches a signal encoding and decoding system such as the signal decoding system shown in FIG. 3, the bit stream received at the decoding system input is a digital audio signal represented in the frequency domain. This input is supplied to inverse quantizer-decoder 4, where it is decoded. The output of inverse quantizer-decoder 4 is fed to inverse discrete transform processor 5, where its inverse discrete transform is returned to the time domain; i.e. the inverse discrete cosine transform

(IDCT), inverse discrete Fourier transform (IDFT), or inverse Karhunen-Loeve transform (IKLT), etc., as applicable, is transformed. The output of inverse discrete transform processor 5 is inverse-windowed by frame buffer 6, and output as a decoded digital audio signal represented in the time domain. The inverse windowing process multiplies each frame of the signal by the inverse of the function used to window it, thereby restoring the amplitude of the audio signal to its original state removing the window components (Kizuki Col. 2 lines 17-34).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Oh in view of Chen to incorporate audio signal is encoded in the frequency domain using one of a plurality of encoding schemes, the method further comprising frequency-domain decoding of the encoded original audio signal as taught by Kizuki to allow for an accurate method of getting information back and forth from the frequency/time domain, wherein window components can be removed and the signal content preserved in the original format (Kizuki Col. 2 lines 17-34).

Re claims 3, 8, and 13, Oh in view of Chen fails to teach the method according to claim 2 wherein said decoding comprises:

decoding said encoded signal using a decoding scheme corresponding to said one of a plurality of encoding schemes; applying an inverse transform to the encoded audio signal;

and applying an inverse window function.

Kizuki teaches a signal encoding and decoding system such as the signal decoding system shown in FIG. 3, the bit stream received at the decoding system input is a digital audio signal represented in the frequency domain. This input is supplied to inverse quantizer-decoder 4, where it is decoded. The output of inverse quantizer-decoder 4 is fed to inverse discrete transform processor 5, where its inverse discrete transform is returned to the time domain; i.e. the inverse discrete cosine transform (IDCT), inverse discrete Fourier transform (IDFT), or inverse Karhunen-Loeve transform (IKLT), etc., as applicable, is transformed. The output of inverse discrete transform processor 5 is inverse-windowed by frame buffer 6, and output as a decoded digital audio signal represented in the time domain. The inverse windowing process multiplies each frame of the signal by the inverse of the function used to window it, thereby restoring the amplitude of the audio signal to its original state removing the window components (Kizuki Col. 2 lines 17-34).

Therefore, it would have been obvious to one of ordinary skill in the art at the time of the invention to modify the system of Oh in view of Chen to incorporate decoding said encoded signal using a decoding scheme corresponding to said one of a plurality of encoding schemes; applying an inverse transform to the encoded audio signal and applying an inverse window function as taught by Kizuki to allow for an accurate method of getting information back and forth from the frequency/time domain, wherein window components can be removed and the signal content preserved in the original format (Kizuki Col. 2 lines 17-34).

Conclusion

5. **THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Michael C. Colucci whose telephone number is (571)-270-1847. The examiner can normally be reached on 9:30 am - 6:00 pm, Monday-Friday.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richemond Dorvil can be reached on (571)-272-7602. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

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